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(71)Applicant : VICTOR CO OF JAPAN LTD
(72)Inventor : NAKASO JIRO
UDAGAWA TOMOYUKI

(54) **AUDIO SIGNAL TRANSMISSION CIRCUIT**

(57)Abstract:
PROBLEM TO BE SOLVED: To provide a coefficient arithmetic unit for a convolver to obtain a coefficient of the convolver provided to the audio signal transmission circuit synchronizing signal in which distortion of a sound image of a high sound frequency especially having been unavoidable by a speaker and a headphone is eliminated to allow the user to enjoy a natural audio signal.
SOLUTION: Changeover devices 7L, 7R are controlled to actually measure a response characteristic of speakers 4L, 4R thereby obtaining a correction filter coefficient of convolvers 2L, 2R and the obtained coefficient is given to the convolvers 2L, 2R, in which convolution is conducted. Thus, in the case of controlling correction of the speakers, an impulse response f0(t) with a specific characteristic depends on the speaker characteristic and the filter coefficient of the convolvers 2L, 2R is selected so that a prescribed medium frequency band has a flat characteristic and only peak levels at a higher frequency band are suppressed.

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CLAIMS

[Claim(s)]
[Claim 1] The convolver which collapses and carries out data processing of the signal from a sound source, and outputs it according to the set-up multiplier, Using the predetermined system of measurement containing a transducer, give the audio signal from a predetermined sound source to said transducer, and it carries out based on impulse response wave h (t) beforehand measured in the measuring point near [said] the transducer. Impulse response f0 of the specific property of said impulse response wave h (t) in said measuring point In the predetermined mid-range of (t) The audio signal propagation circuit which has a multiplier supply means by which only the level more than the predetermined level in a band higher than it supplies the multiplier for inverse filters computed by predetermined level by control processing having been carried out to said convolver by the band of the one lower than a predetermined band becoming flat.

DETAILED DESCRIPTION

[Detailed Description of the Invention]
[0001]
[Field of the Invention] About an audio signal propagation circuit, this invention amends the response characteristic of the loudspeaker of hi-fi audio equipment, and, more specifically, relates to the multiplier arithmetic unit of the convolver which asks

for the multiplier of the convolver prepared in the audio signal propagation circuit reproducing the faithful Hara tone quality.

[0002]

[Description of the Prior Art] Conventionally, by using a digital filter in a loudspeaker system, flattening of the sound pressure and the group delay frequency characteristics of a loudspeaker system is carried out, the response characteristic of a loudspeaker is amended, and a device which reproduces the faithful Hara tone quality is performed.

[0003]

[Problem(s) to be Solved by the Invention] By the way, it faces performing flattening of the sound pressure and the group delay frequency characteristics of the loudspeaker system, and the following three points are mainly mentioned as a factor which obstructs it.

1) Diffraction of a loudspeaker cabinet, and a reflective **** factor.

2) The factor by the phase rotation by the analog filter.

3) The factor by the partial vibration of the diaphragm (a surrounding support system is included) in a loudspeaker unit.

Among these, about the factor of the above 1 and 2, it can amend so satisfactory.

[0004] However, about the factor of 3, it becomes quite difficult amendment. Especially, it tends to generate in the band where a frequency is high, in the case of such a high frequency, partial vibration will arise according to the oscillation mode of each proper of a loudspeaker cone, and a complicated vibration of a wave etc. will produce the partial vibration used as this factor of 3. It is difficult to suppress such a complicated vibration by only equalizing sound pressure level. So, there was a trouble referred to as that playback of a high fidelity sound source is not obtained more for a viewer at a quantity region side. Then, this invention will offer the audio signal propagation circuit which solved such a trouble.

[0005]

[Means for Solving the Problem] This invention is accomplished based on the following fundamental view. It is thought that it is solvable by being the band where the band which is vibrating with a peak and a DIP in fact although it is ideal to use the band where, as for a loudspeaker, the diaphragm is vibrating as one also must be used in many cases, it is hard to produce partial vibration about the factor of 1 and 2 of the above first, and the whole cone vibrates, and equalizing a peak and a DIP.

[0006] Next, since the partial vibration mentioned by the above 3 more than said band which poses a problem arises, on the whole, a loudspeaker cone vibrates, and even if it only equalizes a peak and a DIP like said solution of 1 and 2, a rather complicated wave may be produced from it being thought that it originated in each proper oscillation and the wave has moreover arisen. Therefore, in this band, it is the view of only the sound pressure more than average level being stopped, the sound pressure below average level being changing into a condition as it is, and generating of a

complicated wave being prevented, and making it flat level substantially.
[0007] The convolver which collapses and carries out data processing of the signal from a sound source, and outputs it as the solution means according to the set-up multiplier, Using the predetermined system of measurement containing a transducer, give the audio signal from a predetermined sound source to said transducer, and it carries out based on impulse response wave h (t) beforehand measured in the measuring point near [said] the transducer. Impulse response f0 of the specific property of said impulse response wave h (t) in said measuring point In the predetermined mid-range of (t) The band of the one lower than a predetermined band becomes flat, and the audio signal propagation circuit which has a multiplier supply means by which only the level more than the predetermined level in a band higher than it supplies the multiplier for inverse filters computed by predetermined level by control processing having been carried out to said convolver is offered.
[0008]
[Embodiment of the Invention] A desirable example explains about the gestalt of operation of this invention. Drawing 1 is the block diagram showing one example of the audio signal propagation circuit of this invention. In this drawing, the sound sources 1L and 1R of a two-channel stereo system are the predetermined audio signal sources. The convolvers 2L and 2R for amending the response characteristic of a loudspeaker and amending the amplitude and a phase characteristic to coincidence and Amplifier 3L and 3R are formed between Loudspeakers 4L and 4R and sound sources 1L and 1R.
[0009] It is a control section controlled to amend a loudspeaker by giving the amendment filter coefficient for which it asked to convolver 2L and 2R, collapsing it, and calculating it while 15 carries out change-over control of the change-over machine mentioned later, surveys the response characteristic of Loudspeakers 4L and 4R and asks for the amendment filter coefficient of a convolver.
[0010] The memory 16 for memorizing the amendment filter factor for which it asked based on the observation of the response characteristic of Loudspeakers 4L and 4R is formed. Based on control of a control section 15, Convolvers 2L and 2R are separated for the response characteristic of Loudspeakers 4L and 4R from the transmission system way of an audio signal at the time of an observation. At the time of amendment of the response characteristic of Loudspeakers 4L and 4R, the change-over machines 7L and 7R for switching so that Convolvers 2L and 2R may be formed in the transmission system way of an audio signal are formed.
[0011] Namely, the configuration shown in drawing 1 measures the impulse response of Loudspeakers 4L and 4R. By carrying out computing control of the filter factor of the convolvers 2L and 2R prepared in the transmission system way of an audio signal that the property of Loudspeakers 4L and 4R should be negated, and it should amend in a flat property Amend the amplitude and a phase to coincidence, keep only the predetermined band of a transmission characteristic constant, aim at an improvement

of tone quality, and the faithful Hara tone quality is reproduced. For example, the distortion of the image avoided neither by the loudspeaker nor headphone is removed, and it enables it to enjoy a natural audio signal.
[0012] Here, the filter coefficient of Convolvers 2L and 2R is calculated as multiplier data with the gaging system shown in drawing 2 . Namely, drawing 2 is in the condition of connecting the change-over machines 7L and 7R to the Terminal ga and ha side, respectively, and not forming Convolvers 2L and 2R in drawing 1 . With the microphone 8 formed in the measuring point which is equivalent to a listening location in the anechoic chamber which is not illustrated Measure the impulse response of the loudspeakers 4L and 4R in this location, and the response characteristic of Loudspeakers 4L and 4R is negated. The filter factor of the convolvers 2L and 2R prepared in the transmission system way of an audio signal is computed that it should amend in a flat property. It is a system configuration Fig. for realizing the ideal impulse response which amends the amplitude and a phase characteristic by carrying out a convolution operation and amending the response characteristic of Loudspeakers 4L and 4R.
[0013] The digital I/O board on which 11 sends out the ideal impulse as digital data in drawing 2 , The D/A converter to which 6 carries out D/A conversion of the output of the digital I/O board 11, The amplifier which 7 amplifies the changed signal and is inputted into loudspeaker 4L (or 4R), The microphone which incorporates the signal with which 8 was outputted from loudspeaker 4L (or 4R), the amplifier which amplifies the signal which incorporated 9 with the microphone 8, and 10 are the A/D converters which carry out A/D conversion of the magnification output. The output from above-mentioned A/D converter 10 is incorporated as an impulse response to a workstation 13 through the digital I/O board 11 and a computer 12. Property measurement of loudspeaker 4L (or 4R) before applying amendment is performed, and the operation output of the filter coefficient is carried out as multiplier data based on the measured impulse response wave. In addition, the property of a microphone 8 is amended in process of an operation if needed.
[0014] Namely, the I/O board 11 constitutes a measurement-signal generating means to generate a measurement signal. The configuration of the path of D/A converter 6, amplifier 7, loudspeaker 4L, a microphone 8 - a workstation 13 constitutes a response characteristic measurement means to search for the amplitude characteristic and the phase characteristic which are the response of an audio signal transmission system based on a measurement signal. Moreover, further While a workstation 13 determines the target property replaced so that the amplitude of a predetermined band might be equalized among the response characteristics searched for An operation means to ask for the filter factor of the convolver prepared in the audio signal transmission system so that it might be completed as said target property by the response of an audio signal transmission system is constituted. The system which amends in the property which negated the property of a loudspeaker and carried out flattening

substantially, and aims at a tone-quality improvement is realized.
[0015] Measurement of the impulse response of the loudspeakers 4L and 4R by the configuration shown in drawing 2 is measured in an anechoic chamber using a microphone 8, for example, 1000 synchronous addition is suppressed and measured in a deed error using 4096 times of samples. Drawing 3 shows impulse response wave h (t) obtained by this gaging system, and drawing 4 (a) is the wave which shows the amplitude characteristic which carried out the Fourier transform of the impulse response wave h (t), and drawing 4 (b) shows the wave of the impulse response f0 of the specific property used as a target (t).
[0016] Here, first, in case the workstation 13 shown in drawing 2 asks for a filter factor, to the amplitude characteristic before the amendment shown by drawing 4 (a), it is made into the amendment band which should equalize the inside band of about 8000 substantially from 150Hz, and makes the band which cannot be amended low-pass and the high regions other than this band.
[0017] And the inside of the aforementioned amendment band, The so-called level of the piston band (band which vibrates on the whole according to sound pressure, without accompanying a loudspeaker cone by the wave) which originates in 1 and 2 which pointed out the about 1600Hz frequency band by the term of the above-mentioned trouble, and is produced from 150Hz is equalized. The band exceeding this is recognized as the band which a wave produces on the loudspeaker cone pointed out as the above-mentioned trouble 3, is set up as a target property that the level more than the average level in this band is held down to average level, and computes a filter factor.
[0018] Namely, impulse response f0 of the specific property shown in drawing 4 (b) which is the amplitude characteristic amended based on the measurement property (amplitude characteristic concerning impulse response wave h (t)) (t) It asks. impulse response f0 of the above-mentioned specific property (t) The above-mentioned impulse response wave h (t) from -- the augmented matrix H obtained and its transposed matrix HT f0 Matrix F0 which made (t) one train HT HG=HT F0 It is referred to as filter factor [of the convolvers 2L and 2R which show each element of Determinant G which consists of one train to fulfill to drawing 1] g (n).
[0019] The solution of a top Noriyuki train type is described below. By this example, a response waveform is uniquely obtained on a time-axis by calculating the solution with which a determinant is filled by the above-mentioned configuration. A filter factor which specifically makes the minimum the square of the difference of the impulse response obtained by the input edge and outgoing end of a convolver according to the least square method (bibliography: "a guide to application of a digital filter" and Acoustical Society of Japan 43-volume No. 4 (1987), Haruo Hamada) using the Levinson algorithm shall be obtained.
[0020] the discrete multiplier of the impulse response of now and a convolver -- g1, g2, ..., gm-1 ** -- if it carries out -- the discrete response f0 in a microphone

location, f1, and ... f_{n+m-2} It can express with a degree type.

[0021]

[Equation 1]

[0022] however, hi transfer characteristics and p -- p= -- 0, 1, ..., n+m −2. Formula (1) It is [0023] when it expresses in procession.

[Equation 2]

[0024] A next door and formula (2) It can be further expressed as F=HG. Here, it is the impulse F0 of an input. If the square of the difference of impulse response F in a microphone location is taken and it is a performance index P, it is P=(F−F0) T (F−F0).

=(HG−F0)T (HG−F0)

=(GT HT −F0T)(HG−F0)

= It is [0025] in order to ask for impulse response G of a convolver to be set to GT HT HG−F0 THG−GT HT F0+F0TF0, and for a performance index P serve as the minimum.

[Equation 3]

[0026] It calculates. However, it expresses that T is a transposed matrix. And formula (4) = from 0 HT HG=HT F0 (5) What is necessary is just to determine the solution G which becomes. That is, it is an upper type (5) about a filter factor. It amends so that the level more than average level may be held [in / the amplitude / in / a transmission characteristic is amended by setting up like and / a microphone location / and a phase characteristic are equalized in a piston band, and / a band higher than it] down to average level, and it does not amend as a band which cannot amend low-pass [other than these], and a high region, but considers as the same property as an actual impulse response wave. thus, the distortion of the image which reproduced the faithful Hara tone quality and was avoided neither by the loudspeaker nor headphone -- being removable -- moreover, loudspeaker equimeasure actual in low-pass and the

frequency bands of a high region other than a mid-range -- a law -- the same property as a system is acquired and the natural audio signal fitted in consideration of the property of an actual loudspeaker can be enjoyed to coincidence.
[0027] Drawing 5 is a flow chart which shows the control action which sets up a filter coefficient by workstation 13 using the gaging system shown in drawing 2 . First, with the gaging system shown in drawing 2 , the response of a loudspeaker is measured (step S1), the Fourier transform (FFT: frequency-time-axis conversion) of the this impulse response wave h (t) is carried out, and the amplitude characteristic is acquired (step S2). Next, in step S3, while determining low-pass and the high region which are not amended, the level of each point of these bands is determined.
[0028] On the other hand, step S4 is asked for the average level of ** and a piston band. Here, equalization of all the points is sufficient and equalization of arbitration spacing is sufficient. Next, this equalization level, for example, 88dB, is set up as level of a piston band (step S5).
[0029] And the band used as ka>1 conjectured that a wave arises is set up. That is, the frequency to the frequency set to ka=1 from a frequency (a standard is an inertia control field) sufficiently higher than low-pass resonance is set up.
[0030] Next, in step S7, average level is compared with the conversion result after the measurement obtained in step S2, if the direction of the level of a conversion result is smallness, it will progress to step S9 as it is, and if the level of a conversion result is size, it will progress to step S8 and will be set as the level, i.e., above-mentioned average level, of each point.
[0031] In step S9, low-pass [outside an amendment band] and each point data of a high region, and the amendment data performed in step S8 are compounded, and reverse frequency conversion (IFFT) of this compounded data is carried out (step S9, 10). And finally, in step S10, with the least square method, a target frequency / test frequency is calculated and the multiplier of a convolver is obtained.
[0032] Thus, in this example, the sound source of a loud-sound region can reappear faithfully especially by carrying out speech processing based on the multiplier which can acquire a property which was described above. In addition, in the above-mentioned example, it has amended so that the sound pressure more than average level may be held down to average level in the band exceeding a piston band, but you may amend so that the envelopment property which did not restrict to average level and was set up beforehand may be met. Moreover, although a gaging system is formed in an audio signal propagation circuit system and he is trying to decide a multiplier in the above-mentioned example, the separate gaging system of the above independently is prepared, it asks for the multiplier by the approach beforehand mentioned above, and you may make it store the multiplier in memory 16.
[0033]
[Effect of the Invention] Since he is trying to hold down more than the predetermined level of the sound pressure of high bandwidth which a wave produces on a

loudspeaker cone especially to predetermined level according to the audio signal propagation circuit of this invention as explained above, the sound source of a loud-sound region can reappear faithfully.

DESCRIPTION OF DRAWINGS

[Brief Description of the Drawings]
[Drawing 1] It is the block diagram showing the configuration of one example of the audio signal propagation circuit as the 1st example of this invention.
[Drawing 2] It is the block diagram showing the gaging system of the filter factor of the convolver concerning this invention.
[Drawing 3] It is the explanatory view showing the impulse response wave before amendment.
[Drawing 4] It is the property Fig. showing the amplitude characteristic and the target property which carried out the Fourier transform of the impulse response wave of drawing 3 .
[Drawing 5] It is a flow chart for carrying out filter coefficient calculation by workstation 13 using the gaging system shown in drawing 2 .
[Description of Notations]
2L, 2R Convolver
4L, 4R Loudspeaker
7 Nine Amplifier
7L, 7R Change-over machine
8 Microphone
10 A/D Converter
11 Digital I/O Board (Measurement-Signal Generating Means)
12 Computer
13 Workstation (Operation Means)
15 Control Section (Multiplier Supply Means is Constituted with Memory 16)
16 Memory